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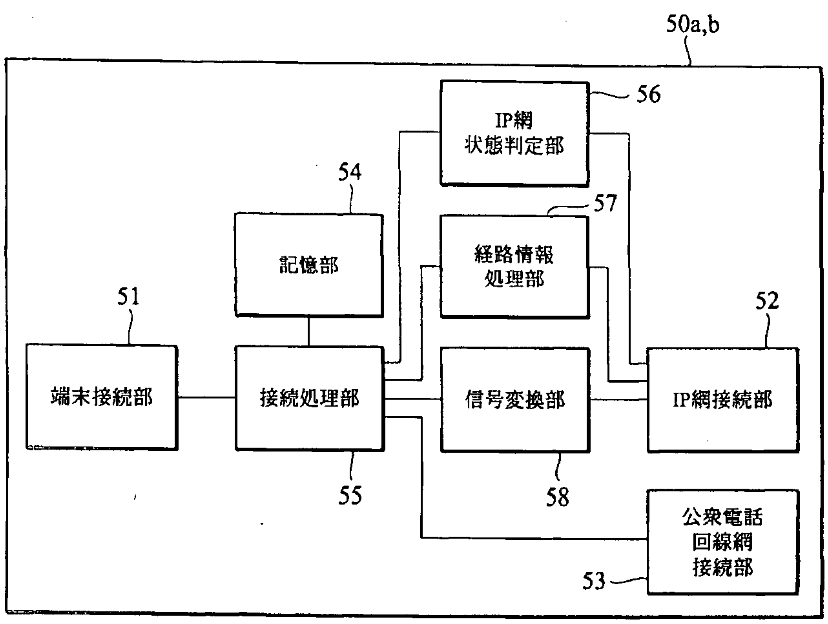
(54) Title: TERMINAL CONNECTION DEVICE, CONNECTION CONTROL DEVICE, AND MULTI-FUNCTION TELEPHONE TERMINAL

(54) 発明の名称: 端末接続装置、接続制御装置及び多機能電話端末



56...IP NETWORK STATE JUDGMENT UNIT
54...STORAGE UNIT
57...PATH INFORMATION PROCESSING UNIT
51...TERMINAL CONNECTION UNIT
55...CONNECTION PROCESSING UNIT
58...SIGNAL CONVERSION UNIT
52...IP NETWORK CONNECTION UNIT
53...PUBLIC TELEPHONE LINE NETWORK CONNECTION UNIT

(57) Abstract: According to a telephone number of call destination contained in a connection request transmitted from a telephone terminal (70a) and information stored in a VoIP gateway (50a), the VoIP gateway (50a) transmits the connection request to an IP network (2) or PSTN (1). When communication

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FULL CONTENTS

[Claim(s)]

[Claim 1]

It is terminal-connection equipment which connects the packet network which sends and receives packet data, the dial-up line network which send and receive a voice band signal, and the communication terminal which transmits the connection request containing the telephone number of a call place, and transmits said connection request for any of said packet network or said dial-up line network being,

The connection processing section which transmits said connection request for any of said packet network or said dial-up line network being based on the telephone number of said call place contained in said connection request,

It has the packet reticulated voice judging section which judges whether communication through said packet network is possible,

It is terminal-connection equipment characterized by transmitting said connection request it was determined that will transmit to said packet network to said dial-up line network when judged with the communication which minded said packet network by said packet reticulated voice judging section being impossible for said connection processing section.

[Claim 2]

Said connection processing section is terminal-connection equipment given in the 1st clause of a claim characterized by transmitting said connection request to said dial-up line network when predetermined identification information is added to the telephone number of said call place contained in said connection request.

[Claim 3]

Said connection processing section is based on said connection request which transmitted to said packet network. The predetermined packet data in which it is shown whether it was set up with the communication terminal to which the telephone number of said call place was assigned for any of the 2nd channel which goes via the 1st channel which goes only via said packet network or said packet network, and said dial-up line network they are is received from said packet network. Terminal-connection equipment given in the 1st clause of a claim characterized by having further the channel information processing section which transmits the signal which reports any should be set as said communication terminal which transmitted said

connection request between said 1st channel or said 2nd channel based on said received predetermined packet data.

[Claim 4]

It is based on the connection request containing the telephone number of the call place transmitted from the terminal-connection equipment connected to the packet network which sends and receives a packet signal, and the dial-up line network which send and receive a voice band signal. It is the connection control unit which sets up the channel of said terminal-connection equipment and the communication terminal to which the telephone number of said call place was assigned, and is installed in said packet screen residue,

As opposed to the communication terminal by which any of said channel which goes via said channel which goes only via said packet network or said packet network, and said dial-up line network were assigned to the telephone number of said call place The connection control unit characterized by having the channel information transmitting section which transmits the predetermined packet data in which it is shown whether it set up to said terminal-connection equipment.

[Claim 5]

Said channel information transmitting section is replaced with said predetermined packet data. As opposed to the communication terminal by which any of said channel which goes via said channel which goes only via said packet network or said packet network, and said dial-up line network were assigned to the telephone number of said call place A connection control unit given in the 4th clause of a claim characterized by transmitting the packet of the voice band signal which shows whether it set up to said terminal-connection equipment.

[Claim 6]

The speech-signal-processing section which performs radial transfer of the voice band signal for voice calls,

The connection processing section which chooses whether said voice call is performed through a dial-up line network, or said voice call is performed through a packet network,

The signal-processing section which performs transform processing between the packet data which can communicate by said voice band signal and said packet screen residue when said voice call was performed through the packet network and it is chosen,

The packet-sending-and-receiving section which sends and receives said packet data from which said voice band signal was changed between said packet networks through the digital subscriber line for un-

The multi-function telephone terminal characterized by preparation ******.

[Claim 7]

Said connection processing section is a multi-function telephone terminal given in the 6th clause of a claim characterized by choosing performing said voice call through a dial-up line network when it is impossible to perform said voice call through said packet network.

[Claim 8]

The base station section which can communicate between PHS terminals as a PHS base station is provided,

Said speech-signal-processing section is a multi-function telephone terminal given in the 6th clause of a claim characterized by performing radial transfer of said voice band signal between said PHS terminals through said base station section.

[Claim 9]
The main phone section which can communicate between cordless phone units as a cordless main phone is provided,
Said speech-signal-processing section is a multi-function telephone terminal given in the 6th clause of a claim characterized by performing radial transfer of said voice band signal between said cordless phone units through said main phone section.
[Claim 10]
The slot which can insert the communication card which enables communication between Personal Digital Assistants by wireless LAN communication is provided,
Said packet-sending-and-receiving section is a multi-function telephone terminal given in the 6th clause of a claim characterized by sending and receiving the packet data which communicated between said Personal Digital Assistants through said communication card inserted in said slot between said packet networks through said asymmetrical digital subscriber line.
[Claim 11]
The video telephone processing section which performs radial transfer of the video signal for a video telephone is provided,
Said video telephone processing section performs transform processing between the packet data which can communicate by said video signal and said packet screen residue,
Said packet-sending-and-receiving section is a multi-function telephone terminal given in the 6th clause of a claim characterized by sending and receiving the packet data which communicated between said video telephone processing means between said packet networks through said asymmetrical digital subscriber line.

[Detailed Description of the Invention]
Technical field
This invention relates to the terminal-connection equipment, connection control unit, and multi-function telephone terminal for providing the call service by a packet network.
Background art
In connection with progress of an information technology (IT) in recent years, it adds to an old subscription call service (Plain Old Telephone System, POTS), and is Voice. over The IP phone service which adopted IP (VoIP) technology is provided.
The packet network for providing IP phone service in such a situation (IP network), By connecting a communication terminal, i.e., the analog telephone terminal used from the former, to the terminal-connection equipment connected to the dial-up line network (PSTN) for providing a subscription call service, and what is called the VoIP gateway The communication terminal concerned can perform communication with other communication terminals via a packet network or a dial-up line network.
Moreover, since the packet network and the dial-up line network are connected through the gateway equipment which performs predetermined signal transformation, the communication terminal concerned can

communicate with other communication terminals connected only to the dial-up line network.
However, the user of the communication terminal concerned checked whether it would be possible to communicate through the communication terminal and packet network of a call place, and there was a problem that it had to direct whether to communicate through a packet network or a dial-up line network to terminal-connection equipment.
Then, this invention was made in view of such a point, and it determines whether to communicate via which network of a packet network or a dial-up line network based on the telephone number of a call place. When a packet network cannot be used, while performing communication which goes via a dial-up line network, it sets it as the purpose to offer the terminal-connection equipment, connection control unit, and multi-function telephone terminal which can make a user recognize the path of a channel with the communication terminal of a call place.
The indication of invention
The packet network with which the 1st feature of this invention sends and receives packet data, and the dial-up line network which send and receive a voice band signal, It is terminal-connection equipment which connects the communication terminal which transmits the connection request containing the telephone number of a call place, and transmits a connection request for any of a packet network or a dial-up line network being. The connection processing section which transmits a connection request for any of a packet network or a dial-up line network being based on the telephone number of the call place contained in the connection request, It has the packet reticulated voice judging section which judges whether communication through a packet network is possible. When the connection processing section is judged as the communication which minded the packet network by the packet reticulated voice judging section being impossible, let it be a summary to transmit the connection request it was determined that will transmit to a packet network to a dial-up line network.
In order to determine whether terminal-connection equipment transmits a connection request for any of a packet network or a dial-up line network being based on the telephone number of the call place contained in the connection request according to this feature, It is avoidable that the user of a communication terminal determines whether it should communicate via which network of a packet network or a dial-up line network. When judged with the communication which minded the packet network by the packet reticulated voice judging section being impossible according to this feature Since the connection processing section transmits the connection request it was determined that will transmit to a packet network to a dial-up line network, the user can communicate without being conscious of whether a packet network is usable.
In the 1st feature of this invention, the 2nd feature of this invention makes it a summary to transmit a connection request to a dial-up line network, when predetermined identification information is added to the telephone number of the call place by which the connection processing section is contained in the connection request.
When predetermined identification information, for example, the information which the specific number in which discernment from the telephone number is possible follows, is added to the telephone number of the call place according to this feature Since a connection request is transmitted to a dial-up line network, the user can choose a packet network or a dial-up line network arbitrarily and easily.
The 3rd feature of this invention is based on the connection request which the connection processing

section transmitted to the packet network in the 1st feature of this invention. The predetermined packet data which any of the 2nd channel which goes via the 1st channel or packet network which goes only via a packet network, and a dial-up line network show whether it was set up with the communication terminal to which the telephone number of the call place was assigned is received from a packet network. Let it be a summary to have further the channel information processing section which transmits the signal which reports whether it was set as the communication terminal which transmitted the connection request based on the received predetermined packet data any of the 1st channel or the 2nd channel they are.

The 4th feature of this invention is based on the connection request containing the telephone number of the call place transmitted from the terminal-connection equipment connected to the packet network which sends and receives a packet signal, and the dial-up line network which send and receive a voice band signal. The channel of terminal-connection equipment and the communication terminal to which the telephone number of the call place was assigned is set up. It is the connection control unit currently installed in packet screen residue. Let it be a summary to have the channel information transmitting section which transmits the predetermined packet data in which it is shown any of the channel which goes via the channel or packet network which goes only via a packet network, and a dial-up line network were set up to the communication terminal assigned to the telephone number of the call place to terminal-connection equipment.

In the 4th feature of this invention, the 5th feature of this invention [the channel information transmitting section] It replaces with predetermined packet data and transmission is carried out for the packet of the voice band signal which shows any of the channel which goes via the channel or packet network which goes only via a packet network, and a dial-up line network were set up to the communication terminal assigned to the telephone number of the call place to terminal-connection equipment with a summary.

According to the 3rd or the 5th feature of this invention, the signal which reports any of the channel which goes via the channel or packet network which goes only via a packet network, and a dial-up line network were set up with the communication terminal to which the telephone number of the call place was assigned is transmitted to the communication terminal which transmitted the connection request. Therefore, the user can recognize easily any should be set up with the communication terminal to which the telephone number of the call place was assigned to the connection request which transmitted between the channel which goes only via a packet network, or the channel which goes via a packet network and a dial-up line network.

The speech-signal-processing section in which the 6th feature of this invention performs radial transfer of the voice band signal for voice calls, The connection processing section which chooses whether a voice call is performed through a dial-up line network, or a voice call is performed through a packet network, When the voice call was performed through the packet network and it is chosen, through the signal-processing section which performs transform processing between the packet data which can communicate by a voice band signal and packet screen residue, and the digital subscriber line for un-between packet networks Let it be a summary to have the packet-sending-and-receiving section which sends and receives the packet data from which the voice band signal was changed.

[according to this feature / the connection processing section / perform / through a dial-up line network / a voice call] Or it chooses whether a voice call is performed through a packet network, and the packet-sending-and-receiving section sends and receives the packet data from which the voice band signal was changed between IP networks through the digital subscriber line (ADSL) for un-. For this reason, only by

connecting a multi-function telephone terminal to a general telephone line (subscriber line) While the IP phone using broadband communications becomes possible and being able to avoid complicated connection and wiring, it is avoidable that a user determines whether it should communicate via which network of a packet network or a dial-up line network.

In the 6th feature of this invention, the 7th feature of this invention makes it a summary to choose performing a voice call through a dial-up line network, when the connection processing section is unable to perform a voice call through a packet network.

Since according to this feature it is connectable with the telephone terminal (for example, a multi-function telephone terminal and a general telephone terminal) of the call place concerned through a dial-up line network even if it is the case where the communication terminal of the call place is not connected to the packet network, The user can communicate without being conscious of whether a packet network is usable.

The 8th feature of this invention possesses the base station section which can communicate between PHS terminals as a PHS (Personal Handyphone System) base station in the 6th feature of this invention. The speech-signal-processing section makes it a summary to perform radial transfer of a voice band signal between PHS terminals through the base station section.

In the 6th feature of this invention, the 9th feature of this invention possesses the main phone section which can communicate between cordless phone units as a cordless main phone, and makes it a summary for the speech-signal-processing section to perform radial transfer of a voice band signal between cordless phone units through the main phone section.

The 10th feature of this invention possesses the slot which can insert the communication card which enables communication between Personal Digital Assistants by wireless LAN communication in the 6th feature of this invention. The packet-sending-and-receiving section makes it a summary to send and receive the packet data which communicated between Personal Digital Assistants through the communication card inserted in the slot between packet networks through an asymmetrical digital subscriber line.

According to this feature, the packet-sending-and-receiving section through an asymmetrical digital subscriber line between packet networks A continuous connection Internet service can be provided to a Personal Digital Assistant, without needing complicated connection and wiring, in order to send and receive the packet data which communicated between Personal Digital Assistants through the communication card inserted in the slot.

The 11th feature of this invention possesses the video telephone processing section which performs radial transfer of the video signal for a video telephone in the 6th feature of this invention. The video telephone processing section performs transform processing between the packet data which can communicate by a video signal and packet screen residue, and the packet-sending-and-receiving section makes it a summary to send and receive the packet data which communicated between the video telephone processing sections between packet networks through an asymmetrical digital subscriber line.

The best form for inventing

[The 1st embodiment]

(Network configuration containing terminal-connection equipment and a connection control unit)

The terminal-connection equipment and the connection control unit concerning one embodiment of this invention are explained referring to Drawings.

Drawing 1 is the figure showing the outline of the network containing the terminal-connection equipment and the connection control unit concerning this embodiment. As shown in drawing 1 , the VoIP gateway 50a (terminal-connection equipment) is connected to the dial-up line network 1 (the following, PSTN1) which is a network for providing a subscription call service (POTS), and IP network 2 which is a network for providing IP phone service. Moreover, the telephone terminal 70a (communication terminal) which is an analog telephone terminal which sends and receives a voice band signal is connected to the VoIP gateway 50a. The VoIP gateway 50a performs conversion with a voice band signal and an IP packet concerned while sending and receiving the telephone terminal 70a and a voice band signal. Moreover, the VoIP gateway 50a sends and receives IP network 2 and an IP packet for PSTN1 and a voice band signal.
Furthermore, the VoIP gateway 50a concerning this embodiment can transmit the connection request concerned for any of PSTN1 or IP network 2 being based on the connection request containing the telephone number of the call place transmitted from the telephone terminal 70a.
[in addition, the VoIP gateway 50a, and PSTN1 and IP network 2] For example, it is connectable using the splitter which performs multiplex [of the xDSL (Digital Subscriber Line) modem which realizes digital transmission on the metallic subscriber's loop, and a voice band signal and a xDSL modem signal / separation and multiplex]. Moreover, the VoIP gateway 50a can also build in a xDSL modem.
On the other hand, while the VoIP gateway 50b is also connected with PSTN1 and IP network 2, the telephone terminal 70b is connected to the VoIP gateway 50b. The VoIP gateway 50b has the same function as the VoIP gateway 50a mentioned above.
L3SW21a and 21b which are the layer 3 switch in which IP network 2 sends and receives an IP packet in this embodiment, It is constituted by the gateway 12 which performs conversion with the packet data sent and received in the voice band signal and IP network 2 which are sent and received in PSTN1, and the call agent 20 (connection control unit) who sets up the channel between telephone terminals.
The call agent 20 sets up the channel between telephone terminals based on the connection request transmitted from the VoIP gateway 50a or the VoIP gateway 50b. [the call agent 20 concerning this embodiment] It has the function to operate as MGC (Media Gateway Controller) specified in MGCP (Media Gateway Control Protocol). Based on MGCP, processing about a setup of the VoIP gateway 50a (50b) or the gateway 12, and a channel etc. is performed.
Moreover, in this embodiment, while the telephone terminal 71 which is a digital telephone terminal is connected, PSTN1 and IP network 2 are connected to PSTN1 through the gateway 12.
In the network configuration explained above, the channel from which a path differs can be set up between the telephone terminal 70a and the telephone terminal 70b. That is, the channel of the telephone terminal 70a and the telephone terminal 70b is set up by any of the path shown below they are.
(a) VoIP gateway 50 a-L3SW21a-IP network 2-L3SW21 b-VoIP gateway 50b
(b) VoIP gateway 50 a-L3SW21a-IP network 2-gateway 12-PSTN1-VoIP gateway 50b
(c) VoIP gateway 50 a-PSTN1-VoIP gateway 50b
In this embodiment, the VoIP gateway 50a (50b) and the call agent 20 set up the channel by the path shown in (the channel which goes via IP network 2 as much as possible, i.e., (a), and b). However, when the obstacle has occurred in a part of IP network 2 (for example, layer 3 switch), the channel which goes via IP network 2 and PSTN1 through the gateway 12 is set up. Moreover, when the user of the telephone terminal

70a (70b) specifies PSTN1 by adding predetermined information to the telephone number, the channel which goes only via PSTN1 sets up the VoIP gateway 50a.
Furthermore, the channel with the telephone terminal 71 connected with the telephone terminal 70a or the telephone terminal 70b only PSTN1 is set up by any of the path shown below they are.
(a) VoIP gateway 50a(50b)-L3SW21a(21b)-IP network 2-gateway 12-PSTN1
(b) VoIP gateway 50a(50b)-PSTN1
Also about the channel of the telephone terminal 70a or the telephone terminal 70b, and the telephone terminal 71, the VoIP gateway 50a (50b) and the call agent 20 set up the channel by the channel which goes via IP network 2 as much as possible, i.e., the path shown in (a). However, about communication using the special service (for example, action-addressee accounting service) which cannot be processed through IP network 2, the channel by the path shown in (b) is set up.
(Composition of terminal-connection equipment)
Next, with reference to drawing 2 , the functional-block composition of the terminal-connection equipment 50a concerning this embodiment, i.e., the VoIP gateway, is explained. In addition, since the VoIP gateway 50b also has the same composition as the VoIP gateway 50a as mentioned above, the composition of the VoIP gateway 50a is explained hereafter.
As shown in drawing 2 , the VoIP gateway 50a has the terminal-connection section 51, the IP network terminal area 52, the dial-up line network (PSTN) terminal area 53, the storage section 54, the connection processing section 55, the IP network state judging section 56, the channel information processing section 57, and the signal transformation section 58.
The terminal-connection section 51 offers Interface Division for connecting the telephone terminal 70a. Specifically, the terminal-connection section 51 sends and receives the telephone terminal 70a, a voice band signal, etc. using FXS (Foreign Exchange Station) Interface Division which has an RJ-11 connector. Moreover, the terminal-connection section 51 can perform supplying power required for the operation of the telephone terminal 70a etc.
The IP network terminal area 52 offers Interface Division for connecting the subscriber's loop with IP network 2, and possesses LAN Interface Division, such as 100 BASE-TX. For example, the IP network terminal area 52 is connected with the metallic subscriber's loop through a xDSL modem. Moreover, as mentioned above, when the VoIP gateway 50a builds in a xDSL modem, the IP network terminal area 52 also has the function of a xDSL modem. In this case, the IP network terminal area 52 possesses the RJ-11 connector for sending and receiving a xDSL modem signal, and is connected to the metallic subscriber's loop.
The PSTN terminal area 53 offers Interface Division for connecting the subscriber's loop of PSTN1, and possesses FXO (Foreign Exchange Office) Interface Division which has an RJ-11 connector. For example, the PSTN terminal area 53 is connected with the metallic circuit concerned through a splitter, when the IP network terminal area 52 is connected to the metallic subscriber's loop through the xDSL modem.
The storage section 54 memorizes the information which matched telephone number [of a call place], IP network 2, or PSTN1. Specifically, the storage section 54 memorizes the information about the telephone number of a call place with required performing communication which goes via PSTN1 as an "PSTN detour table." The storage section 54 can transmit compulsorily the connection request to the special service (for

example, action-addressee accounting service) which cannot be processed through IP network 2, the high communication of urgency to the police etc., etc. to PSTN1 by memorizing the information concerned.

Drawing 3 is the figure showing the example of the "PSTN detour table" memorized by the storage section 54. For example, as shown in the 1st line of an "PSTN detour table", when the single figure of the telephone number of the call place contained in the connection request transmitted from the telephone terminal 70a is "1", it is judged with the connection request which should transmit to PSTN1 by the connection processing section 55 mentioned later. Moreover, also when the telephone number of a call place starts in "003O", "003**", etc. similarly, it is judged with the connection request which should transmit to PSTN1. [redacted]

The connection processing section 55 transmits the connection request concerned for any of IP network 2 or PSTN1 being based on the information memorized by the telephone number and the storage section 54 of the call place contained in the connection request transmitted from the telephone terminal 70a. Moreover, when judged with the communication which minded IP network 2 by the IP network state judging section 56 being impossible for the connection processing section 55, the connection request concerned it was determined that will transmit to IP network 2 is transmitted to PSTN1. Furthermore, the connection processing section 55 can transmit the connection request concerned to PSTN1 compulsorily, when predetermined identification information is added to the telephone number of the call place.

The connection processing section 55 acquires the telephone number of the call place contained in the connection request transmitted from the telephone terminal 70a, and, specifically, collates the acquired telephone number based on the "PSTN detour table" memorized by the storage section 54. When the telephone number concerned corresponds to the conditions shown using the information about the telephone number memorized as an "PSTN detour table", the connection processing section 55 transmits the connection request concerned to PSTN1.

When the connection processing section 55 does not correspond to the conditions shown using the information about the telephone number the telephone number concerned is remembered to be as an "PSTN detour table" on the other hand, The connection request concerned is transmitted to IP network 2 through signal transformation section 58 grade using the NTFY (Notification) command specified in MGCP. Moreover, when four "0" follows the head of the telephone number of the call place contained in the connection request transmitted from the telephone terminal 70a in this embodiment and the connection processing section 55 is added to it, The telephone number concerned transmits the connection request concerned to PSTN1 irrespective of whether it corresponds to the conditions shown using the information about the telephone number memorized as an "PSTN detour table."

Therefore, the user of the telephone terminal 70a can specify PSTN1 arbitrarily and easily at every communication by adding four "0" to the head of the telephone number of a call place.

Furthermore, the connection processing section 55 can receive the result of a judgment of whether communication through IP network 2 is possible from the IP network state judging section 56. When judged with communication through IP network 2 being impossible for the connection processing section 55, the

connection request concerned judged that transmits to IP network 2 based on the telephone number of a call place is also transmitted to PSTN1. The user of the telephone terminal 70a can communicate by operating, as the connection processing section 55 mentioned above, without being conscious of whether IP network 2 is usable.

In addition to the processing of transmission of a connection request mentioned above, the connection processing section 55 processes the connection request which received. [the section] specifically while the connection processing section 55 processes the connection request by the CRCX (Create Connection) command transmitted from IP network 2 The voice band signal changed from packet data is transmitted to the telephone terminal 70a through the terminal-connection section 51 by the signal transformation section 58. Moreover, the connection processing section 55 transmits the voice band signal received through the PSTN terminal area 53 through the terminal-connection section 51 to the telephone terminal 70a while processing the connection request which received from PSTN1.

The IP network state judging section 56 judges whether communication through IP network 2 is possible, and constitutes the packet reticulated voice judging section from this embodiment.

The IP network state judging section 56 specifically minds the subscriber's loop connected to the IP network terminal area 52. While transmitting an ICMP(Internet Control Message Protocol) Echo request packet to L3SW21a, the response from L3SW21a to the Echo request packet concerned is supervised. It judges with communication through IP network 2 being impossible for the IP network state judging section 56, when an Echo request packet is periodically transmitted to L3SW21a and there is no predetermined response. In addition, the judgment method of the state of concrete IP network 2 which the IP network state judging section 56 depends is mentioned later. Moreover, in this embodiment, in order to judge whether communication through IP network 2 is possible, ICMP is used, but you may use network administration protocols, such as SNMP (Simple Network Management Protocol).

The channel information processing section 57 is based on the connection request which the connection processing section 55 transmitted to IP network 2. The predetermined packet data in which it is shown whether it was set up with the telephone terminal to which the telephone number of the call place was assigned for any of the channel which goes via the channel or IP network 2 which goes only via IP network 2, and PSTN1 they are is received from IP network 2. Moreover, the channel information processing section 57 transmits the signal which reports any of the channel which goes via the channel or IP network 2 which goes only via IP network 2, and PSTN1 were set as the telephone terminal 70a which transmitted the connection request concerned based on the received packet data concerned.

[the channel information processing section 57] specifically if the MDCX (modify connection) command of the contents specified beforehand is received from the call agent 20 The audible signal which has a predetermined ON/OFF pattern is transmitted to the telephone terminal 70a through the connection processing section 55 and the terminal-connection section 51.

Drawing 4 shows the example of a pattern of the signal transmitted to the telephone terminal 70a according to reception of the MDCX command concerned. If the MDCX command concerned is received once, the channel information processing section 57 will transmit once the signal which has -25dBm outgoing level to the telephone terminal 70a using the frequency of 400Hz, as shown in drawing 4 . Therefore, the user of the telephone terminal 70a can hear the audible sound "PU PU PU", if the signal shown in drawing 4 is

outputted with the telephone terminal 70a. In addition, the concrete notice method of the information about the path of a channel using the MDCX command and the signal concerned concerned is mentioned later.
The signal transformation section 58 performs conversion with a voice band signal and packet data between the IP network terminal area 52 and the connection processing section 55. Specifically, the signal transformation section 58 is ITU-T. While providing CODEC based on G.729 a/b etc., conversion with the voice band signal and packet data which were digitized by the CODEC concerned can be performed. Furthermore, the signal transformation section 58 can add and remove the RTP (Real-time Transport Protocol) header used in order that reservation etc. may carry out real time nature of an IP packet.
(Composition of a connection control unit)
Next, with reference to drawing 5 , the composition of the connection control unit 20 concerning this embodiment, i.e., a call agent, is explained.
As shown in drawing 5 , the call agent 20 has 20d of storage sections, and 20f of channel information transmitting sections and the setup information transmitting section 20e. [the receiving section 20a the channel setting section 20b, the database retrieval section 20c, and]
The receiving section 20a receives the connection request containing the telephone number of a call place from the VoIP gateway 50a or the VoIP gateway 50b. Specifically, the receiving section 20a receives the response (ACK) to the NTFY command transmitted based on MGCP, the RQNT (Request Notification) command which the setup information transmitting section 20e transmitted, etc. from the VoIP gateway 50a (50b). Furthermore, the receiving section 20a transmits the contents of the command and the response which received to the channel setting section 20b.
20d of storage sections associate and memorize the telephone number currently assigned to the telephone terminal 70a and the telephone terminal 70b and the IP address currently assigned to the VoIP gateway 50a, the VoIP gateway 50b, and the gateway 12.
The database retrieval section 20c retrieves the information on the telephone number memorized by 20d of storage sections, or an IP address based on the directions from the channel setting section 20b, and notifies the result of search to the channel setting section 20b.
[the section] while the channel setting section 20b acquires the IP address related with the telephone number of a call place from the database retrieval section 20c based on the contents of the command transmitted from the receiving section 20a Information, including an IP address required for a setup of a channel, a port number, the profile of RTP, etc., is sent and received using SDP (Session Description Protocol).
For example, when the channel setting section 20b sets up the channel of the telephone terminal 70a and the telephone terminal 70b, Based on the connection request transmitted from the VoIP gateway 50a, information, including the telephone number of the telephone terminal 70b, the IP address of the VoIP gateway 50a, the port number to be used, the profile of RTP, etc., is transmitted to the VoIP gateway 50b. Here, when the VoIP gateway 50b does not answer within predetermined time, it judges with a setup of the channel which goes only via IP network 2 being impossible for the channel setting section 20b, and the information concerned is transmitted to the gateway 12. The gateway 12 transmits required information, including the telephone number of the telephone terminal 70b etc., to PSTN1 based on the information concerned transmitted through the setup information transmitting section 20e from the channel setting

section 20b. Moreover, when the telephone terminal to which the telephone number of the call place was assigned is connected only to PSTN1 (this embodiment telephone terminal 71), the channel setting section 20b transmits the information concerned to the gateway 12 similarly.

Thus, it is set up any of the channel which goes via the channel or IP network 2 which goes only via IP network 2 between the telephone terminal 70a and the telephone terminal 70b, and PSTN1 they are.

20f of channel information transmitting sections transmit the predetermined packet data in which it is shown any of the channel which goes via the channel or IP network 2 which goes only via IP network 2, and PSTN1 were set up to the telephone terminal assigned to the telephone number of the call place to the VoIP gateway 50a (50b).

Specifically, 20f of channel information transmitting sections receive the information which shows any of the channel which goes via the channel or IP network 2 which goes only via IP network 2, and PSTN1 were set up from the channel setting section 20b. 20f of channel information transmitting sections transmit the MDCX packet beforehand specified to the VoIP gateway 50a (50b) which transmitted the connection request based on the received information concerned. In this embodiment, 20f of channel information transmitting sections transmit the MDCX packet concerned to the VoIP gateway 50a (50b) twice, when the channel which goes only via IP network 2 is set up. Moreover, 20f of channel information transmitting sections transmit the MDCX packet concerned to the VoIP gateway 50a (50b) once, when the channel which goes via IP network 2 and PSTN1 is set up.

Moreover, 20f of channel information transmitting sections are replaced with the MDCX packet mentioned above. You may transmit the packet data of a voice band signal in which it is shown any should be set up to the communication terminal assigned to the telephone number of said call place between the channel which goes only via IP network 2, or said channel which goes via IP network 2 and PSTN1 to the VoIP gateway 50a (50b).

Specifically, 20f of channel information transmitting sections can transmit twice the IP packet equivalent to the voice band signal shown in drawing 4 to the VoIP gateway 50a (50b), when the channel which goes only via IP network 2 is set up. Moreover, 20f of channel information transmitting sections can transmit once the IP packet equivalent to the voice band signal shown in drawing 4 to the VoIP gateway 50a (50b), when the channel which goes via IP network 2 and PSTN1 is set up.

(Procedure of the connection request by terminal-connection equipment)

Next, with reference to drawing 6 , the procedure of the connection request by the terminal-connection equipment 50a (50b) concerning this embodiment, i.e., the VoIP gateway, is explained. In addition, since the VoIP gateway 50b also has the same composition as the VoIP gateway 50a as mentioned above, operation of the VoIP gateway 50a is explained hereafter.

First, if the user of the telephone terminal 70a does off-hook [of the telephone terminal 70a] and dials the telephone number of a call place, the VoIP gateway 50a will receive the connection request in which the telephone number concerned transmitted with the telephone terminal 70a is contained (S11). The VoIP gateway 50a acquires the telephone number data concerned, and memorizes it temporarily in the connection processing section 55 (S12).

Subsequently, the VoIP gateway 50a refers to the "PSTN detour table" memorized by the storage section 54 (S13). The telephone number concerned memorized in the contents and the connection processing section

55 of the "PSTN detour table" is collated, and it is judged whether the connection request concerned should be transmitted to PSTN1 (S14).

In Step S14, the telephone number concerned memorized by the connection processing section 55 does not correspond to the conditions shown by the contents of the "PSTN detour table". When judged with transmitting the connection request concerned to IP network 2, the VoIP gateway 50a judges whether it is ready-for-sending ability for the connection request concerned to IP network 2 based on the judgment of the use propriety of IP network 2 by the IP network state judging section 56 (S15).

In Step S15, the VoIP gateway 50a changes the connection request concerned into an IP packet (NTFY command), when it judges that the connection request concerned transmits to IP network 2 (S16). Furthermore, the VoIP gateway 50a transmits the IP packet concerned to IP network 2 (S17).

The NTFY command concerned transmitted to IP network 2 in Step S17 is processed by the call agent 20, and the channel of the telephone terminal 70a and the telephone terminal with which the telephone number of the call place concerned is given is set up after that.

On the other hand, in Step S14, when the VoIP gateway 50a judges that the connection request concerned transmits to PSTN1, and when it judges that IP network 2 is [use] impossible in Step S15, the VoIP gateway 50a transmits the connection request concerned to PSTN1 (S18).

Next, with reference to drawing 7 , the judgment method of the state of IP network 2 by the IP network state judging section 56 of the VoIP gateway 50a concerning this embodiment is explained.

First, as for the IP network state judging section 56, the telephone terminal 70a checks whether it is in a state on hook or an off-hook state (S21). The IP network state judging section 56 judges with the telephone terminal 70a not being used when the telephone terminal 70a is in a state on hook, and when it is in an off-hook state, the telephone terminal 70a judges it to be under use (S22).

The IP network state judging section 56 is ICMP to L3SW21a to which the VoIP gateway 50a is connected through the subscriber's loop in Step S22 when the telephone terminal 70a is judged to be a state on hook. An Echo packet is transmitted 4 times (S23). Subsequently, the IP network state judging section 56 is ICMP from L3SW21a. ICMP which is the response to an Echo packet Echo It is checked whether a Reply packet is received once or more (S24).

In Step S24, it is ICMP from L3SW21a. Echo When a Reply packet is received once or more, the IP network state judging section 56 performs different processing by whether it had judged that IP network 2 is usable until it starts processing of Step S22 (S25).

That is, when having judged that IP network 2 is usable, the IP network state judging section 56 judges that IP network 2 is usable succeedingly, until it starts processing of Step S22 ("YES" in S25, and S28).

On the other hand, when having judged with it being impossible using IP network 2, the IP network state judging section 56 reports that the VoIP gateway 50a became usable about IP network 2 to the call agent 20, until it starts processing of Step S22 (S26). Subsequently, the IP network state judging section 56 checks the existence of the response from the call agent 20 to the notice concerned (S27). In Step S27, when there is a response from the call agent 20, the IP network state judging section 56 judges that IP network 2 is usable (S28).

Moreover, in Step S27, when there is no response from the call agent 20, it judges with the IP network state judging section 56 being impossible using IP network 2 (S29). In addition, in Step S24, the IP network state

judging section 56 is ICMP from L3SW21a. Echo The IP network state judging section 56 also judges the case where a Reply packet is not received at all with it being impossible using IP network 2.
Then, the IP network state judging section 56 stands by for 15 seconds from the processing in Step S28 or Step S29, and repeats the processing which begins from Step S21 and which was mentioned above (S30).
Moreover, in Step S22, when the telephone terminal 70a is judged to be an off-hook state, the IP network state judging section 56 stands by for 15 seconds from the processing in Step S22 similarly, and repeats the processing which begins from Step S21 and which was mentioned above.
in addition, the thing for which the IP network state judging section 56 concerning this embodiment sends and receives a predetermined ICMP packet between the VoIP gateway 50a and L3SW21a -- the state of IP network 2 -- judging -- although -- It replaces with ICMP and is IETF. It is good also as a form which sends and receives the SNMP request specified by RFC1901 grade, and a response between the VoIP gateway 50a and L3SW21a. Moreover, the VoIP gateway 50a is good also as other network devices (un-illustrating) which replace with L3SW21a and constitute IP network 2, and a form which send and receive the packet concerned, a request, or a response among the call agents 20.
(Communication procedure using terminal-connection equipment and a connection control unit)
Next, with reference to drawing 8 and drawing 9 , the communication procedure using the terminal-connection equipment concerning this embodiment, the connection control unit 50a, i.e., the VoIP gateway, the VoIP gateway 50b, and the call agent 20 is explained.
Drawing 8 shows the communication procedure in case the channel which goes via IP network 2 is set up between the telephone terminal 70a, the telephone terminal 70b 50a, i.e., the VoIP gateway, and the VoIP gateway 50b. In addition, this communication procedure shows the case where it judges that IP network 2 is usable, by the IP network state judging section 56 of the VoIP gateway 50a mentioned above.
First, in order to perform communication with the telephone terminal 70b, the telephone terminal 70a is off-hook (S41). The VoIP gateway 50a connected with the telephone terminal 70a transmits the NTFY command which shows that the telephone terminal 70a was off-hook to IP network 2 (S42a). IP network 2 transmits the response (return code 200) to the NTFY command concerned to the VoIP gateway 50a (S42b).
Furthermore, IP network 2 transmits the RQNT (Request Notification) command to the VoIP gateway 50a in order to make information, including the telephone number of a call place etc., transmit to the VoIP gateway 50a (S43).
The VoIP gateway 50a which received the RQNT command transmits the dial tone which shows that sending out of the call place telephone number is possible to the telephone terminal 70a (S44). Moreover, the VoIP gateway 50a transmits the response to the RQNT command concerned to IP network 2 (S45).
Then, the telephone terminal 70a transmits the connection request containing the telephone number (call place telephone number) of the telephone terminal 70b (S46).
The VoIP gateway 50a which received the connection request concerned transmits the NTFY command containing the telephone number of the call place contained in the connection request concerned to IP network 2 (S47a). IP network 2 transmits the response to the NTFY command concerned to the VoIP gateway 50a (S47b).
Furthermore, the CRCX command for which what the preparation which receives an IP address, a port

number to be used, etc. of the VoIP gateway 50a which is needed for a setup of a channel was able to carry out as for IP network 2 is shown is transmitted to the VoIP gateway 50a (S48a). [the VoIP gateway 50a which received the CRCX command concerned] The information, including a port number, a RTP profile, etc., used for communication of the direction of VoIP gateway 50b is transmitted to IP network 2 based on SDP from the IP address of the VoIP gateway 50a, and the VoIP gateway 50a (S48b).

Subsequently, IP network 2 transmits the information transmitted by the VoIP gateway 50a as a CRCX command to the VoIP gateway 50b (S49a). The VoIP gateway 50b which received the CRCX command concerned transmits the information, including a port number, a RTP profile, etc., used for communication of the direction of VoIP gateway 50a to IP network 2 based on SDP from the VoIP gateway 50b (S49b).

IP network 2 which received the information concerned from the VoIP gateway 50b transmits the predetermined MDCX command (the MDCX command in inactive mode) to the VoIP gateway 50a (S50a). a response of as opposed to the MDCX command concerned in the VoIP gateway 50a -- IP network 2 -- transmitting (S50b) -- the audible signal which has the predetermined ON/OFF pattern shown in drawing 4 is transmitted to the telephone terminal 70a (S51). Furthermore, IP network 2 and the VoIP gateway 50a repeat the same processing as Step S51 from step S50a (from S52a to S53).

Thus, IP network 2 transmits the predetermined MDCX command to the VoIP gateway 50a twice, when the channel which goes only via IP network 2 is set up between the telephone terminal 70a and the telephone terminal 70b (S50a and S52a). Thereby from the telephone terminal 70a, the audible signal which has the predetermined ON/OFF pattern shown in drawing 4 is outputted twice.

Here, when the audible sound concerned is outputted twice from the telephone terminal 70a, it shall be well-known to it being shown that the channel which goes only via IP network 2 was set up to the user beforehand.

Therefore, the user of the telephone terminal 70a can recognize that the channel which goes only via IP network 2 was set up between the telephone terminal 70a and the telephone terminal 70b, when audible sound, such as "PU PU PU and PU PU PU", is heard.

Subsequently, IP network 2 transmits the information based on SDP transmitted in step S49b from the VoIP gateway 50b to the VoIP gateway 50a (S54a). The VoIP gateway 50a receives the information concerned, and transmits the response of the purport that communicative was ready to IP network 2 (S54b).

IP network 2 which received the response from the VoIP gateway 50a transmits the RQNT command which directs the call of the telephone terminal 70b to the VoIP gateway 50b (S55a). the VoIP gateway 50b which received the RQNT command concerned -- the telephone terminal 70b -- calling (S55b) -- the response to the RQNT command concerned is transmitted to IP network 2 (S55c).

IP network 2 which received the response from the VoIP gateway 50b transmits the MDCX command which shows that the VoIP gateway 50b is calling the telephone terminal 70b to the VoIP gateway 50a (S56a). the VoIP gateway 50a which received the MDCX command concerned -- a ring back tone -- the telephone terminal 70a -- transmitting (S56b) -- the response to the MDCX command concerned is transmitted to IP network 2 (S56c).

Then, the user of the telephone terminal 70b answers the call in step S55b, and the VoIP gateway 50b transmits the NTFY command for which it is shown that the telephone terminal 70b was off-hook to IP network 2 (S57). When processing of Step S57 is completed, communication is started using the channel

which goes via IP network 2 between the telephone terminal 70a and the telephone terminal 70b.
In addition, [in this embodiment, the audible signal which notifies the telephone terminal 70a that the channel which goes via IP network 2 was set up is transmitted, before a ring back tone is transmitted (above-mentioned step S56b) (above-mentioned steps S51 and S53), but] The sending-out timing of the audible signal concerned and the notice timing of the predetermined MDCX command (above-mentioned step S50a and S52a) may be the times (above-mentioned step S57) of a telephone call being started between the telephone terminal 70a and the telephone terminal 70b etc.
Moreover, it may replace with transmission of the predetermined MDCX command (above-mentioned step S50a and S52a), and the IP packet equivalent to the audible signal (voice band signal) shown in drawing 4 may be transmitted from IP network 2. In this case, the call agent 20 installed on IP network 2 transmits the IP packet concerned to the telephone terminal 70a.
Drawing 9 shows the communication procedure in case the channel which goes via IP network 2 and PSTN1 is set up between the telephone terminals 71 connected with the telephone terminal 70a only PSTN1.
In addition, this communication procedure shows the case where it judges that IP network 2 is usable, like the communication procedure shown in drawing 8 by the IP network state judging section 56 of the VoIP gateway 50a. Moreover, since Step S61 shown in drawing 9 to step S68b supports step S48b from Step S41 shown in drawing 8 , respectively, it omits the explanation and explains the processing after step S69a.
[gateway / 50a / VoIP / the NTFY command with which the telephone number of the telephone terminal 71 is contained / IP network 2 which received in step S67a] It judges with setting up the channel which goes via IP network 2 and PSTN1 through the gateway 12 based on the telephone number concerned, and a predetermined MDCX packet is transmitted to the VoIP gateway 50a (S69a). a response of as opposed to the MDCX packet concerned in the VoIP gateway 50a -- IP network 2 -- transmitting (S69b) -- the audible signal which has the predetermined ON/OFF pattern shown in drawing 4 is transmitted to the telephone terminal 70a (S70).
Thus, IP network 2 transmits a predetermined MDCX packet to the VoIP gateway 50a once, when the channel which goes via IP network 2 and PSTN1 is set up between the telephone terminal 70a and the telephone terminal 71. Thereby from the telephone terminal 70a, the audible signal which has the predetermined ON/OFF pattern shown in drawing 4 is outputted once.
Here, when the audible sound concerned is outputted once from the telephone terminal 70a, it shall be well-known to it being shown that the channel which goes via IP network 2 and PSTN1 was set up to the user beforehand.
Therefore, the user of the telephone terminal 70a can recognize that the channel which goes via IP network 2 and PSTN1 was set up between the telephone terminal 70a and the telephone terminal 71, when the audible sound "PU PU PU" is heard.
Subsequently, IP network 2 transmits information, including the IP address of the gateway 12, the port number to be used, a RTP profile, etc., to the VoIP gateway 50a based on SDP (S71a). The VoIP gateway 50a receives the information concerned, and transmits the response of the purport that communicative was ready to IP network 2 (S71b).
IP network 2 which received the response from the VoIP gateway 50a transmits the message (SET UP

message) of a connection request to the telephone terminal 71 through the gateway 12 (S72a). The telephone terminal 71 is CALL which shows that it is during processing of the connection request concerned. A PROC (Call Processing) message is transmitted to IP network 2 (S72b).
Moreover, the telephone terminal 71 transmits the ALERT message which shows that the output of ringing tone was started in the call 71 of the user of the telephone terminal 71, i.e., a telephone terminal, to IP network 2 (S73).
IP network 2 which received the ALERT message transmits the MDCX message for making a ring back tone transmit to the VoIP gateway 50a (S74a). a response of as opposed to the MDCX message concerned in the VoIP gateway 50a which received the MDCX message concerned -- IP network 2 -- transmitting (S74b) -- a ring back tone is transmitted to the telephone terminal 70a (S75).
Then, the user of the telephone terminal 71 answers the call started in Step S73, and the telephone terminal 71 transmits the CONN (Connect) message which shows that the telephone terminal 71 was off-hook to IP network 2 (S76a).
IP network 2 which received the CONN message is CONN which shows the check to a CONN message. An ACK (Connect Acknowledge) message is transmitted to the telephone terminal 71 (S76b).
When processing of step S76b is completed, communication is started using the channel which goes via IP network 2 and PSTN1 between the telephone terminal 70a and the telephone terminal 71.
In addition, [in this embodiment, the audible signal which notifies the telephone terminal 70a that the channel which goes via IP network 2 and PSTN1 was set up is transmitted, before a ring back tone is transmitted (above-mentioned step S75) (above-mentioned step 70), but] The sending-out timing of the audible signal concerned and the notice timing of the predetermined MDCX command (above-mentioned step S69a) may be the times (above-mentioned step S76b) of a telephone call being started between the telephone terminal 70a and the telephone terminal 71 etc.
Moreover, it may replace with transmission of the predetermined MDCX command (above-mentioned step S69a), and the IP packet equivalent to the audible signal (voice band signal) shown in drawing 4 may be transmitted from IP network 2. In this case, the call agent 20 installed on IP network 2 transmits the IP packet concerned to the telephone terminal 70a.
As mentioned above, although the case where MGCP was used was explained as an example about one embodiment of this invention, of course, this invention can also be realized using other connection control protocols (Session Initiation Protocol), for example, SIP.
(The operation and effect by the terminal-connection equipment by this embodiment, and a connection control unit)
According to this embodiment, it is based on the information memorized by the telephone number and the storage section 54 of the call place contained in the connection request. Since it determines whether the VoIP gateway 50a (50b) transmits a connection request for any of IP network 2 and PSTN1 being, it is avoidable that the user of the telephone terminal 70a (70b) determines whether it should communicate via IP network 2 or which network of PSTN1.
When judged with the communication which minded IP network 2 by the IP network state judging section 56 being impossible according to this embodiment Since the connection processing section 55 transmits the connection request it was determined that will transmit to IP network 2 to PSTN1, the user can communicate

without being conscious of whether IP network 2 is usable.

Since a connection request is transmitted to PSTN1 when predetermined identification information, for example, the information which the specific number in which discernment from the telephone number is possible follows, is added to the telephone number of the call place according to this embodiment, the user can choose IP network 2 or PSTN1 arbitrarily and easily.

According to this embodiment, the audible signal which reports any of the channel which goes via the channel or IP network 2 which goes only via IP network 2, and PSTN1 were set up with the telephone terminal to which the telephone number of the call place was assigned is transmitted to the telephone terminal which transmitted the connection request. [when communication which is followed for example, in which the telephone terminal of a call place goes via IP network 2 according to an obstacle etc. cannot be performed, or when the telephone terminal of the call place is connected only with PSTN1] It can be reported to a user that the call agent 20 set up the channel which goes via IP network 2 and PSTN1 through the gateway 12 based on the transmitted connection request.

[The 2nd embodiment]

(The whole communication system composition including a multi-function telephone terminal)

Next, it explains, referring to Drawings about the multi-function telephone terminal concerning one embodiment of this invention. Drawing 10 is the outline block diagram showing communication system including the multi-function telephone terminals 30a and 30b concerning this embodiment, and drawing 11 is the functional block diagram of the multi-function telephone terminals 30a and 30b concerning this embodiment.

As shown in drawing 10 , in communication system including the multi-function telephone terminals 30a and 30b concerning this embodiment, the ISP servers 9 and 10, the gateway 12, and the call control server 13 are installed in IP networks 2, such as the Internet, possible [connection].

IP network 2 is the distributed computer network which connected the global communication network mutually using TCP/IP which is a communications protocol, and transmission and reception of packet data are mutually attained between [various] terminals through TCP/IP.

Moreover, the multi-function telephone terminals 30a and 30b are connected to ISP (Internet service provider) server and PSTN(dial-up line network) 1 through ADSL 3 and 4 and splitters 7 and 8. The multi-function telephone terminals 30a and 30b may be connected to ISP servers 9 and 10 and PSTN1 through high speed lines, such as a CATV circuit and an optical cable, instead of ADSL 7 and 9.

The ISP servers 9 and 10 are server equipment which provides the connection service to IP network 2, and connect a user's various terminals to IP network 2 through the access lines 3 and 4, such as ADSL, here. Moreover, the ISP servers 9 and 10 give the "IP address" which is a peculiar identifier to each multi-function telephone terminals 30a and 30b, when the multi-function telephone terminals 30a and 30b are connected to IP network 2.

The gateway 12 is a telephone call signal converter which connects IP network 2 and PSTN1 mutually, and changes mutually the voice band signal which can communicate in packet data and PSTN1 which can communicate in IP network 2.

The call control server 13 is connected to the administrative database 14 which accumulates the "IP address" which is the identifier given to each multi-function telephone terminals 30a and 30b connected to

IP network 2, and the record which associates the registered "telephone number."
The call control server 13 notifies the "IP address" of the multi-function telephone terminal 30b (or 30a) of the communication destination registered into the administrative database 14 to the inquiry from the multi-function telephone terminal 30a (or 30b).
[in addition, the telephone number registered into the administrative database 14 in this embodiment] It is the telephone number for IP phones, and in order to distinguish from the telephone number of the subscription telephone in PSTN1, it is the form which added "***" to the head part like "***-1234-5678."
Thereby, the call control server 13 can identify whether the user wishes the voice call (IP phone) through an IP network by judging whether "***" is added to the telephone number acquired from the multi-function telephone terminal 30a of the transmitting agency.
Moreover, the call control server 13 possesses the notice section 13a, the general circuit selection section 13b, the database retrieval section 13c, and 13d of Interface Division sections and the communication history Management Department 13e, as shown in drawing 10 .
It connects with the general circuit selection section 13b, and the notice section 13a notifies the information about the gateway 12 chosen by the general circuit selection section 13b to the multi-function telephone terminal 30a of a transmitting agency.
The general circuit selection section 13b is connected to the notice section 13a and the database retrieval section 13c. On the administrative database 14, when the notice of the purport that the "IP address" matched with the "telephone number" of the communication destination transmitted from the multi-function telephone terminal 30a does not exist is received from the database retrieval section 13c, the gateway 12 is chosen based on the "telephone number" concerned. [0162]
Connect with the administrative database 14 and the general circuit selection section 13b, and the database retrieval section 13c minds 13d of Interface Division sections. When the "telephone number" of a telephone call place is acquired, the administrative database 14 is searched and the "IP address" matched with the "telephone number" concerned is detected from the multi-function telephone terminal 30a registered into the administrative database 14. Moreover, the database retrieval section 13c notifies that to the general circuit selection section 13b, when the "IP address" matched with the "telephone number" concerned does not exist.
Moreover, the database retrieval section 13c can attest whether the multi-function telephone terminal 30a can perform a voice call (IP phone) using this communication system. In this case, the database retrieval section 13c attests using the "IP address" of the multi-function telephone terminal 30a, a "MAC address", or the "user ID" and the "password" that were transmitted from the multi-function telephone terminal 30a.
It connects with the database retrieval section 13c and the communication history Management Department 13e, and 13d of Interface Division sections supervise the connection-confirm signal transmitted from each multi-function telephone terminal 30a. Here, 13d of Interface Division sections delete the record including the "IP address" of the multi-function telephone terminal 30a which was not able to check a connection signal from the administrative database 14 through the database retrieval section 13c, when the following connection-confirm signal is not checked, even if it carries out predetermined time progress.
Moreover, the "telephone number" of the communication destination where 13d of Interface Division sections were transmitted from the multi-function telephone terminal 30a, Or information required for

authentication of a an "IP address", a "MAC address", a "user ID", a "password", etc. which were transmitted *from* the multi-function telephone terminal 30a is transmitted to the database retrieval section 13c.

The communication history Management Department 13e is connected to 13d of Interface Division sections. Counting (count) of the data number or the amount of data of packet data which each multi-function telephone terminals 30a and 30b ******(ed) is carried out, and the counting result (a packet data number or packet amount of data) is held as a "communication history." The communication history Management Department 13e can also acquire the packet data number concerned or the packet amount of data from the multi-function telephone terminals 30a and 30b.

Moreover, the multi-function telephone terminals 30a and 30b can carry out counting (count) of the number which accessed the call control server 13, and the communication history Management Department 13e can hold the counting result (access number) as a "communication history." The communication history Management Department 13e can also acquire the access number concerned from the multi-function telephone terminals 30a and 30b.

And the communication history Management Department 13e can give the point based on above-mentioned counting results (a packet data number, the packet amount of data, an access number, etc.) for every agency which sold the multi-function telephone terminals 30a and 30b which performed the voice call. Thereby, the eagerness to sell of the agency which sells the multi-function telephone terminals 30a and 30b can be raised.

The "telephone number" of the user who made the registration process the administrative database 14 in order for this communication system to receive service is registered. It is matched with this registered "telephone number", and the "IP address" of the multi-function telephone terminals 30a and 30b which each user uses is recorded. [0169] [0170]

In addition, in this embodiment, since an IP address is changed by the ISP servers 9 and 10 at any time, it has updated serially the present IP address of the multi-function telephone terminals 30a and 30b based on the acknowledge signal from the multi-function telephone terminals 30a and 30b.

Moreover, the administrative database 14 is recording the information for attesting the multi-function telephone terminals 30a and 30b which can perform a voice call using this communication system, including an "IP address", a "MAC address", a "user ID", a "password", etc.

Moreover, the table for the gateways which matched "area code" of an every place region and the "IP address" of the gateway 12 installed in the every place region is also stored in the administrative database 14 concerning this embodiment.

The multi-function telephone terminals 30a and 30b are ******(ed) between IP networks 2 by using the voice band signal for voice calls as packet data, and it has the function which transmits and receives the packet data concerned to the IP address acquired from the call control server 13 based on the telephone number of a telephone call place. [0173]

Specifically [the multi-function telephone terminals 30a and 30b] As shown in drawing 10 and drawing 11 , a hand set 31 and a display 32, A video camera 33, an antenna 35, and the slot 36 that can insert PC card 34, The terminal (LINE) 41 for the telephone lines, a splitter 42, the packet-sending-and-receiving section 43, the connection processing section 44, the VoIP application 45, the speech-signal-processing section 46, antenna I/F47, and the video telephony application 48 are provided.

The telephone lines 5 and 6 by which the terminal (LINE) 41 for the telephone lines is connected to the modular plug sockets 15 and 16 are inserted. Since use of the multi-function telephone terminals 30a and 30b is attained only by connecting the telephone lines 5 and 6 to the terminal (LINE) 41 for the telephone lines, they can avoid complicated wiring work.

Moreover, the voice band signal and packet data for voice calls are made intermingled, or a splitter 42 makes the voice band signal and packet data for voice calls which are intermingled on ADSL separate in ADSL.

A splitter 42 divides into the voice band signal and packet data for voice calls the signal transmitted from the terminal (LINE) 41 for the telephone lines, transmits the voice band signal for voice calls to the connection processing section 44, and, specifically, transmits packet data to the packet-sending-and-receiving section 43. Moreover, a splitter 42 makes the voice band signal for voice calls transmitted from the connection processing section 44, and the packet data transmitted from the packet-sending-and-receiving section 43 intermingled, and transmits to the terminal (LINE) 41 for the telephone lines.

The packet-sending-and-receiving section 43 constitutes the packet-sending-and-receiving section which sends and receives the packet data from which the voice band signal was changed by the VoIP application 45 between IP networks 2 through ADSL3 in this embodiment.

Moreover, the packet-sending-and-receiving section 43 sends and receives the packet data which communicated between Personal Digital Assistants 61 through the communication cards 34 (PCMCIA card etc.) inserted in the slot 36 between IP networks 2 through ADSL3.

Moreover, the packet-sending-and-receiving section 43 sends and receives the packet data which communicated between the video telephone processing means (video telephony application) 48 between IP networks 2 through ADSL3.

Moreover, the packet-sending-and-receiving section 43 can carry out counting (count) of the data number or the amount of data of packet data which the multi-function telephone terminal 30a ******(ed), and can hold the counting result (a packet data number or packet amount of data) as a "communication history."

Moreover, the packet-sending-and-receiving section 43 is predetermined timing, and can also transmit the packet data number concerned or the packet amount of data to the call control server 13.

Moreover, the multi-function telephone terminal 30a can carry out counting (count) of the number which accessed the call control server 13, and the packet-sending-and-receiving section 43 can hold the counting result (access number) as a "communication history." Moreover, the packet-sending-and-receiving section 43 is predetermined timing, and can also transmit the access number concerned to the call control server 13.

The connection processing section 44 constitutes the connection processing section which chooses whether a voice call is performed through PSTN1, or a voice call is performed through IP network 2 in this embodiment.

When the telephone number of the partner point dialed by the calling party is the telephone number for IP phones, the connection processing section 44 will be chosen if a voice call is performed through IP network 2, and, specifically, connects the speech-signal-processing section 46 and the VoIP application 45. On the other hand, when the telephone number of the partner point dialed by the calling party is the telephone number of a subscription telephone, the connection processing section 44 will be chosen if a voice call is

performed through PSTN1, and connects the speech-signal-processing section 46 and a splitter 42.

Moreover, when it is impossible to perform a voice call through IP network 2, the connection processing section 44 may be constituted so that it may choose performing a voice call through PSTN1. Specifically, the connection processing section 44 chooses performing a voice call through PSTN1, when it is detected that the voice call the VoIP application 45 minded IP network 2 is more nearly impossible than various Reasons.

In this embodiment, the VoIP application 45 constitutes the signal-processing section which performs transform processing between the packet data which can communicate on a voice band signal and IP network 2 concerned, when the voice call was performed through IP network 2 and it is chosen.

After changing the VoIP application 45 into packet data for every predetermined cycle while it digital-data-izes the voice band signal which is an analog signal, specifically, it adds a header etc. according to protocols, such as TCP/IP. [0183] [0185]

Moreover, from the packet data received from the packet-sending-and-receiving section 43, the VoIP application 45 restores the voice band signal which is an analog signal, and transmits to the speech-signal-processing section 46.

Moreover, the VoIP application 45 transmits to IP network 2 through ADSL3 by using as digital data the telephone number of the communication destination dialed by the calling party. Moreover, the VoIP application 45 acquired the IP address of the communication destination from the call control server 13, and has also achieved the function which adds this IP address to the packet data from which the voice band signal was changed as a header.

The VoIP application 45 can also memorize the telephone number, an IP address, etc. which were assigned to the self-opportunity (multi-function telephone terminal 30a).

For example, the VoIP application 45 corresponds to H.323, SIP, and the call control protocol of MGCP/MEGACO (Media Gateway Control Protocol).

The speech-signal-processing section 46 constitutes a speech-signal-processing means to perform radial transfer of the voice band signal for voice calls, in this embodiment. Specifically, the speech-signal-processing section 46 outputs the voice band signal for voice calls transmitted from a splitter 42, the VoIP application 45, or the video telephony application 48 through the loudspeaker of the hand set 31 which the multi-function telephone terminal 30a possesses. Moreover, the speech-signal-processing section 46 transmits the voice band signal for voice calls inputted through the microphone of the hand set 31 which the multi-function telephone terminal 30a possesses to a splitter 42, the VoIP application 45, or the video telephony application 48.

Moreover, the speech-signal-processing section 46 can output voice, beeping, a melody, etc. based on the message signal acquired from the call control server 13 or the gateway 12.

Moreover, the speech-signal-processing section 46 can perform radial transfer of a voice band signal between PHS terminals 63 through antenna I/F47. Specifically, the speech-signal-processing section 46 transmits the voice band signal for voice calls transmitted from PHS terminal 63 to a splitter 42 or the VoIP application 45. Moreover, the speech-signal-processing section 46 transmits the voice band signal for voice calls transmitted from a splitter 42 or the VoIP application 45 to PHS terminal 63 through antenna I/F47.

Moreover, the speech-signal-processing section 46 can perform radial transfer of a voice band signal between the cordless phone units 62 through antenna I/F47. Specifically, the speech-signal-processing

section 46 transmits the voice band signal for voice calls transmitted from the cordless phone unit 62 to a *splitter* 42 or the VoIP application 45. Moreover, the speech-signal-processing section 46 transmits the voice band signal for voice calls transmitted from a splitter 42 or the VoIP application 45 to the cordless phone unit 62 through antenna I/F47.

Antenna I/F47 constitute the base station means which can communicate between PHS terminals 63 as a PHS base station through an antenna 35 in this embodiment. Moreover, antenna I/F47 constitute the main phone means which can communicate between the cordless phone units 62 as a cordless main phone through an antenna 35 in this embodiment. Here, you may prepare separately antenna I/F which constitutes a base station means, and antenna I/F which constitutes a main phone means.

The video telephony application 48 constitutes a video telephone processing means to perform radial transfer of the video signal for a video telephone, in this embodiment. Moreover, the video telephony application 48 performs transform processing between the packet data which can communicate on a video signal and this IP network 2. [0197] [0198] [0199]

Specifically [the video telephony application 48] The video signal including the picture signal acquired with the video camera 33 which the multi-function telephone terminal 30a possesses, and the voice band signal transmitted from the speech-signal-processing section 46 is changed into packet data, and the changed packet data is transmitted to the packet-sending-and-receiving section 43.

Moreover, the video telephony application 48 restores a picture signal and a voice band signal from the [redacted] from the packet-sending [redacted] display 32, and transmits [redacted] band signal to the [redacted] 46.

The communication card 34 inserted in the slot 36 changes the packet data transmitted from the packet-sending-and-receiving section 43 according to a protocol for wireless LAN, a protocol for PHS, etc. of IEEE802.11 grade, and transmits to Personal Digital Assistant 61 by wireless communications. Moreover, the communication card 34 changes the data transmitted from Personal Digital Assistant 61 into the packet data which can communicate by IP network 2, and transmits to the packet-sending-and-receiving section 43.

(Operation of above-mentioned communication system)

Operation of above-mentioned communication system is explained with reference to drawing 12 or drawing 16 .

Drawing 12 is the flow figure showing operation at the time of starting of the multi-function telephone terminal 30a concerning this embodiment, and drawing 13 and drawing 14 are the flow figures showing operation at the time of the voice call using the multi-function telephone terminal 30a concerning this embodiment.

In addition, the case where the multi-function telephone terminal 30b or telephone 11 of a communication destination is telephoned is explained to an example from the multi-function telephone terminal 30a of a communicating agency here.

[redacted]

is sent as packet data towards the [redacted] [0208]

The ISP server 9 which received this acknowledge signal publishes the IP address to the multi-function telephone terminal 30a, and transmits to the multi-function telephone terminal 30a as packet data (S303). The multi-function telephone terminal 30a memorizes the acquired IP address to the VoIP application 45. Subsequently, the multi-function telephone terminal 30a of a transmitting agency transmits the assigned IP address and the telephone number of a self-opportunity to the call control server 13. In the call control server 13 side, about whether the telephone number which received is registered, the administrative database 14 is collated, and when registered, the telephone number and the IP address which were received are associated and it registers with the administrative database 14 (S304).

Here, the call control server 13 can also perform authentication about the ability of voice call service to be provided to the multi-function telephone terminal 30a of a transmitting agency with the "MAC address" and the "user ID" which were transmitted from the multi-function telephone terminal 30a, a "password", etc. [0207]

Moreover, since the multi-function telephone terminal 30a of the transmitting agency accessed the call control server 13 (administrative database 14) at this time, you may update the communication history in the packet-sending-and-receiving section 43 (access number).

In addition, also in the multi-function telephone terminal 30b of a communication destination, the same processing as the above-mentioned steps S301-S304 shall be made, and the telephone number and the IP address of the multi-function telephone terminal 30b of a communication destination shall also be registered into the administrative database 14.

Then, each multi-function telephone terminals 30a and 30b send a connection-confirm signal to the call control server 13 periodically, in order to check that connection with IP network 2 is maintained (S305). This connection-confirm signal is serially supervised in the call control server 13 (S306).

That is, the call control server 13 waits to send ("Y" in Step S306), and the following connection-confirm signal (S305), while checking serially whether the connection-confirm signal from each multi-function telephone terminals 30a and 30b is transmitted periodically and checking being transmitted.

On the other hand, even if it carries out predetermined time progress, when the following connection-confirm signal is not checked, ("N" in Step S306) and the multi-function telephone terminals 30a and 30b concerned judge that it does not connect with IP network 2, and delete the IP address registered into the administrative database 14 (S307).

Only the IP address of the multi-function telephone terminals 30a and 30b connected to present IP network 2 will be accumulated into the administrative database 14 by this.

Subsequently, with reference to drawing 13 , operation at the time of the voice call using a multi-function telephone terminal is explained.

First, the telephone number of a communication destination is dialed in the multi-function telephone terminal 30a. This telephone number adds and dials "***" to the head of the usual telephone number, when the communication destination has joined this system.

In response to this dial control, as for the VoIP application 45, the multi-function telephone terminal 30a judges whether connection is established between IP networks 2 through the telephone line 5, ADSL3, a splitter 7, and an ISP server (S401).

When connection between IP networks 2 is established, the VoIP application 45 transmits the telephone number of the multi-function telephone terminal 30b of the communication destination dialed to the call control server 13 by the calling party (S402).
The call control server 13 searches the administrative database 14 based on the telephone number which received. The IP address of the multi-function telephone terminal 30b of a communication destination is searched (S403), and it is judged whether the multi-function telephone terminal 30b of the communication destination (communications partner) is registered into the administrative database 14 (S404).
When the IP address of the multi-function telephone terminal 30b of a communication destination is detected on the administrative database 14, the detected IP address is returned to the multi-function telephone terminal 30a of a communicating agency. Thereby, the multi-function telephone terminal 30a acquires the IP address of the multi-function telephone terminal 30b of a communication destination (S405).
At this time, the multi-function telephone terminal 30a checks whether transmission and reception of data are possible based on the IP address of the multi-function telephone terminal 30b of a communication destination. That is, the multi-function telephone terminal 30a checks whether the direct communication of a communication destination is possible with a firewall etc. (S406).
Connection will be made to establish if ****** of data is possible. On the other hand, when direct communication cannot do a communication destination by a firewall etc., an access method is acquired from the communication destination side (S409), and connection is made to establish.
After connection is established, the intelligent terminal 30b of a communication destination transmits the message which reports that communication establishment was carried out to the multi-function telephone terminal 30a, and outputs a message by the loudspeaker of a hand set 31 at the multi-function telephone terminal 30a which received this message (S407). Subsequently, the voice call (IP phone) through IP network 2 is started (S408).
When it is judged that the IP address of the multi-function telephone terminal 30b of a communication destination is not detected, and the multi-function telephone terminal 30b of a communication destination is not registered in S404 The call control server 13 analyzes the telephone number dialed by the calling party, and chooses the nearby gateway 12 of the multi-function telephone terminal 30b of a communication destination by area code etc. (S410). Here, the gateway 12 shown in drawing 10 should be chosen. Under the present circumstances, the information about the selected gateway 12 is transmitted to the multi-function telephone terminal 30a of transmitting (for example, area code in which the gateway 12 is installed) origin.
At the multi-function telephone terminal 30a which acquired this transmitted information on the gateway 12, (S411), Based on the area code of the gateway 12 etc., the telephone directory table in the VoIP application 45 is collated, and the communication range in PSTN1 judges whether it becomes a long distance (S412).
For example, when not becoming a long distance, after making the message which tells that output from the loudspeaker of a hand set 31 (S418), it connects through PSTN1 that there is a communication destination in a proximity area etc. to the multi-function telephone terminal 30b of a communication destination, and the usual voice call is started (S419).
In S412, when it is judged that it is a long-distance telephone, the multi-function telephone terminal 30a requires selection of whether to permit connection (S413). When a user permits connection ("YES" in S414) The IP address of the gateway 12 is acquired from the call control server 13 (S415), it connects with the

gateway 12 through IP network 2 (S416), and a voice call with the multi-function telephone terminal 30b *which* is a communication destination is started by PSTN1 by gateway 12 course (S417).
When a user does not permit connection in S414 (connection is stopped in "NO" in S414 (S420).)
In S401, when it is judged that the multi-function telephone terminal 30a is not connected to IP network 2, it changes to the connection processing to PSTN1. [judge whether at this time, the multi-function telephone terminal 30a becomes a long distance from the area code of the dialed telephone etc., as shown in drawing 14 (S501), and] in not being a long distance After outputting the message connected not by IP network 2 course but by PSTN1 (S505), the usual voice call is started (S506).
When it is judged by S501 that it becomes a long distance, the multi-function telephone terminal 30a requires connection permission (S502), and a user's selection is urged to it (S503). When a user grants a permission, after outputting the message which becomes a long-distance telephone by PSTN1 (S505), the usual voice call is started (S506).
In S503, when a user does not grant a permission, connection is stopped (S504).
In addition, although premised on the voice call with the multi-function telephone terminal 30b of a communication destination connectable with IP network 2 in the example mentioned above For example, since it is dialed without adding "***" to the head of the telephone number when telephoning to the telephone 11 which has not been registered into the call control server 13, the usual voice call will be performed through PSTN1.
In S402, the multi-function telephone terminal 30a of a transmitting agency may update the communication history in the packet-sending-and-receiving section 43 (access number), and may update the communication history in the packet-sending-and-receiving section 43 (access number) in S408, and 417 and 419.
Subsequently, with reference to drawing 15 and drawing 16 , the data flow in the procedure at the time of an above-mentioned voice call is explained.
Drawing 15 is the sequence diagram showing a data flow in case the multi-function telephone terminal 30a of an origination side and the multi-function telephone terminal 30b of a destination side are in the state of registering with the call control server 13.
As shown in this figure, a calling party sets to the multi-function telephone terminal 30a of an origination side. When the telephone number of the multi-function telephone terminal 30b of a destination side is dialed, [the VoIP application 45 of the multi-function telephone terminal 30a of an origination side] The telephone number of the multi-function telephone terminal 30b of a destination side is transmitted to the call control server 13 through ADSL3, a splitter 7, and the ISP server 9 (S601).
Based on the telephone number of the multi-function telephone terminal 30b of a destination side, the call control server 13 searches the administrative database 14, and detects the IP address of the multi-function telephone terminal 30b of a destination side (S602).
The call control server 13 transmits the IP address of the multi-function telephone terminal 30b of the detected destination side to the multi-function telephone terminal 30a of an origination side (S603).
The VoIP application 45 of the multi-function telephone terminal 30a of an origination side performs connection processing between the multi-function telephone terminals 30b of a destination side using the IP address of the multi-function telephone terminal 30b of a destination side (S604).

[the VoIP application 45 of the multi-function telephone terminal 30a of an origination side] after the end of connection processing The voice band signal inputted through the microphone of a hand set 31 is changed into packet data, and the IP address of the multi-function telephone terminal 30b of a destination side is added to the header etc., and it transmits to it (S605). Here, the packet-sending-and-receiving section 43 of the multi-function telephone terminal 30a of an origination side updates a communication history (a packet data number or packet amount of data).

The received packet data is changed into the multi-function telephone terminal 30b of a destination side by the voice band signal, it is outputted as a voice band signal from the microphone of a hand set, and a voice call is performed.

Drawing 16 is the sequence diagram showing the data flow in the case of being in the state where the multi-function telephone terminal 30a of the origination side is registered into the call control server 13, and the telephone 11 of the destination side is not registered into the call control server 13.

As shown in this figure, a calling party sets to the multi-function telephone terminal 30a of an origination side. If the telephone number of the telephone 11 of a destination side is dialed, the VoIP application 45 of the multi-function telephone terminal 30a of an origination side will transmit the telephone number of the telephone 11 of a destination side to the call control server 13 through ADSL3, a splitter 7, and the ISP server 9 (S701).

Based on the telephone number of the telephone 11 of a destination side, the call control server 13 searches the administrative database 14. Since the record including the IP address of the telephone 11 of a destination side is deleted from on the administrative database 14 when the telephone 11 of the destination side is not registered into the call control server 13, the IP address concerned is not detected. For this reason, the call control server 13 analyzes the area code of the dialed telephone number etc., and searches the nearby gateway 12 of the telephone 11 of a destination side (S702).

The call control server 13 transmits the IP address of the detected gateway 12 to the multi-function telephone terminal 30a of an origination side (S703).

The VoIP application 45 of the multi-function telephone terminal 30a of an origination side performs connection processing between the gateways 12 using the IP address of the gateway 12 (S704).

[the VoIP application 45 of the multi-function telephone terminal 30a of an origination side] after the end of connection processing The voice band signal inputted through the microphone of a hand set 31 is changed into packet data, and the telephone number of the IP address of the gateway 12 and the telephone 11 of a destination side is added to the header etc., and it transmits to it (S705). Here, the packet-sending-and-receiving section 43 of the multi-function telephone terminal 30a of an origination side updates a communication history (a packet data number or packet amount of data).

The gateway 12 changes the packet data concerned into the signal transmission in which ****** is possible by PSTN1 based on the telephone number of the telephone 11 of the destination side in the received packet data, and is transmitted to the telephone 11 of a destination side through PSTN1 (S706).

The telephone 11 of a destination side outputs the signal transmission which received as a voice band signal from the loudspeaker of a hand set, and a voice call is performed.

(The operation and effect by the multi-function telephone terminal concerning this embodiment)

According to the multi-function telephone terminal concerning this embodiment, the packet-sending-and-

receiving section 43 through ADSL3 between IP networks 2 Since the packet data from which the voice *band* signal was changed is sent and received, only by connecting the telephone line (subscriber line) 5 to the terminal 41 for the telephone lines of the multi-function telephone terminal 30a, the IP phone using broadband communications becomes possible, and complicated connection and wiring can be avoided.
Moreover, since according to the multi-function telephone terminal concerning this embodiment it is connectable with the telephone 11 of a communication destination through PSTN1 even if it is the case where the telephone 11 of the communication destination is not connected to IP network 2, the reliability as communication system can be raised.
Moreover, according to the multi-function telephone terminal concerning this embodiment, the calling party can choose an IP phone automatically and can aim at reduction of communication fee money.
Moreover, according to the multi-function telephone terminal concerning this embodiment, the packet-sending-and-receiving section 43 through ADSL3 between IP networks 2 A continuous connection Internet service can be provided to Personal Digital Assistant 61, without needing complicated connection and wiring, in order to send and receive the packet data which communicated between Personal Digital Assistants 61 through the communication card 34 inserted in the slot 36.
Industrial availability
As explained above, according to this invention, it determines whether to communicate via which network of a packet network or a dial-up line network based on the telephone number of a call place. When a packet network cannot be used, while performing communication which goes via a dial-up line network, it sets it as the purpose to offer the terminal-connection equipment, connection control unit, and multi-function telephone terminal which can make a user recognize the path of a channel with the communication terminal of a call place.
[Brief Description of the Drawings]
Drawing 1 is the figure showing the outline of the network containing the terminal-connection equipment and the connection control unit concerning the 1st embodiment of this invention.
Drawing 2 is the figure showing functional block of the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 3 is the figure showing the example of the information memorized by the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 4 is the figure showing the example of the path notification signal transmitted by the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 5 is the figure showing functional block of the connection control unit concerning the 1st embodiment of this invention.
Drawing 6 is a flow chart which shows the procedure of the connection request by the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 7 is a flow chart which shows operation of the packet reticulated voice judging section of the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 8 is the sequence diagram showing the communication procedure concerning the 1st embodiment of this invention.
Drawing 9 is the sequence diagram showing the communication procedure concerning the 1st embodiment

of this invention.
Drawing 10 is the figure showing the outline of communication system including the multi-function telephone terminal concerning the 2nd embodiment of this invention.
Drawing 11 is the figure showing functional block of the multi-function telephone terminal concerning the 2nd embodiment of this invention.
Drawing 12 is the figure showing the flow of operation at the time of starting of the multi-function telephone terminal concerning the 2nd embodiment of this invention.
Drawing 13 is a flow chart which shows the communication procedure using the multi-function telephone terminal concerning the 2nd embodiment of this invention.
Drawing 14 is a flow chart which shows the communication procedure using the multi-function telephone terminal concerning the 2nd embodiment of this invention.
Drawing 15 is the sequence diagram showing the communication procedure using the multi-function telephone terminal concerning the 2nd embodiment of this invention.
Drawing 16 is the figure showing the sequence at the time of the telephone call using the multi-function telephone terminal concerning the 2nd embodiment of this invention.

[Brief Description of the Drawings]
Drawing 1 is the figure showing the outline of the network containing the terminal-connection equipment and the connection control unit concerning the 1st embodiment of this invention.
Drawing 2 is the figure showing functional block of the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 3 is the figure showing the example of the information memorized by the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 4 is the figure showing the example of the path notification signal transmitted by the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 5 is the figure showing functional block of the connection control unit concerning the 1st embodiment of this invention.
Drawing 6 is a flow chart which shows the procedure of the connection request by the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 7 is a flow chart which shows operation of the packet reticulated voice judging section of the terminal-connection equipment concerning the 1st embodiment of this invention.
Drawing 8 is the sequence diagram showing the communication procedure concerning the 1st embodiment of this invention.
Drawing 9 is the sequence diagram showing the communication procedure concerning the 1st embodiment of this invention.
Drawing 10 is the figure showing the outline of communication system including the multi-function telephone terminal concerning the 2nd embodiment of this invention.
Drawing 11 is the figure showing functional block of the multi-function telephone terminal concerning the 2nd

embodiment of this invention.

Drawing 12 is the figure showing the flow of operation at the time of starting of the multi-function telephone terminal concerning the 2nd embodiment of this invention.

Drawing 13 is a flow chart which shows the communication procedure using the multi-function telephone terminal concerning the 2nd embodiment of this invention.

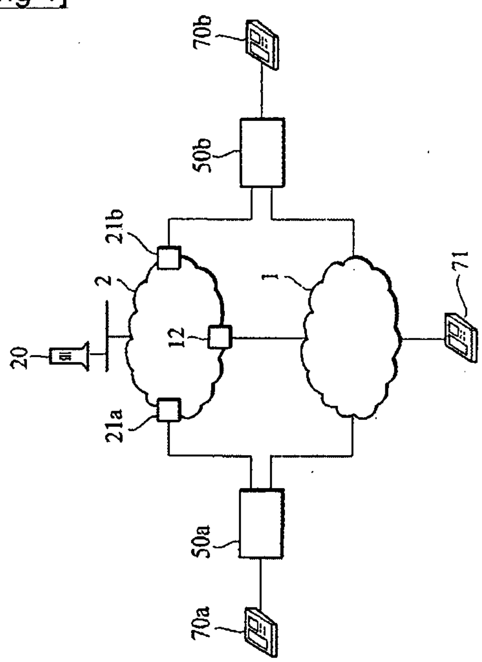
Drawing 14 is a flow chart which shows the communication procedure using the multi-function telephone terminal concerning the 2nd embodiment of this invention.

Drawing 15 is the sequence diagram showing the communication procedure using the multi-function telephone terminal concerning the 2nd embodiment of this invention.

Drawing 16 is the figure showing the sequence at the time of the telephone call using the multi-function telephone terminal concerning the 2nd embodiment of this invention.

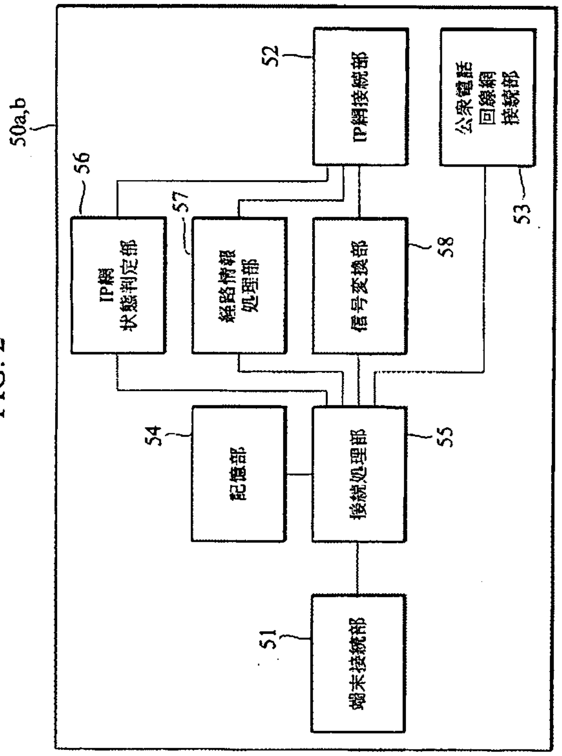
[Drawing 1]

FIG. 1



[Drawing 2]

FIG. 2



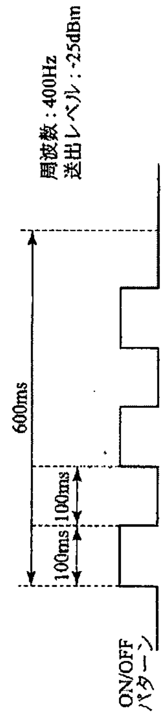
[Drawing 3]

FIG. 3

Digit
1
#
*
003○
003△
⋮
004○
004△
⋮
005○
005△
⋮
010○
⋮

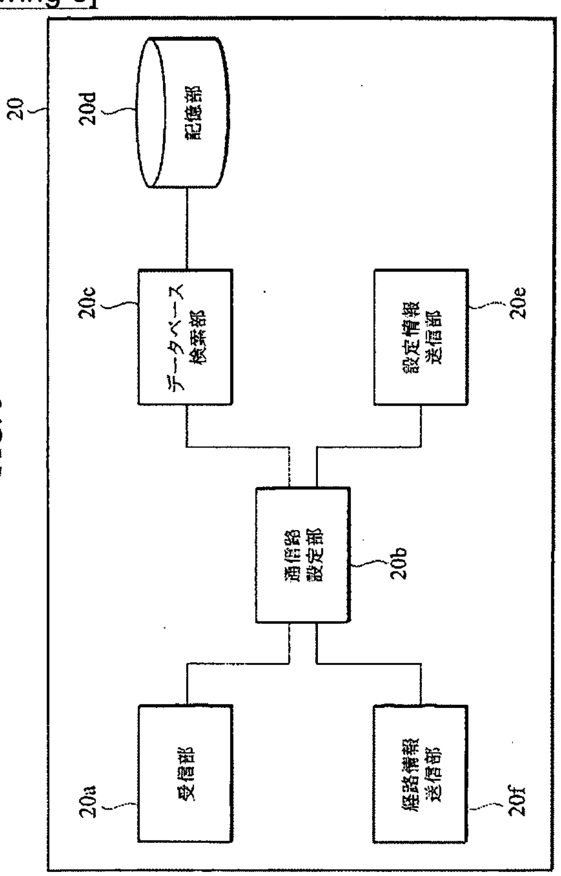
[Drawing 4]

FIG. 4



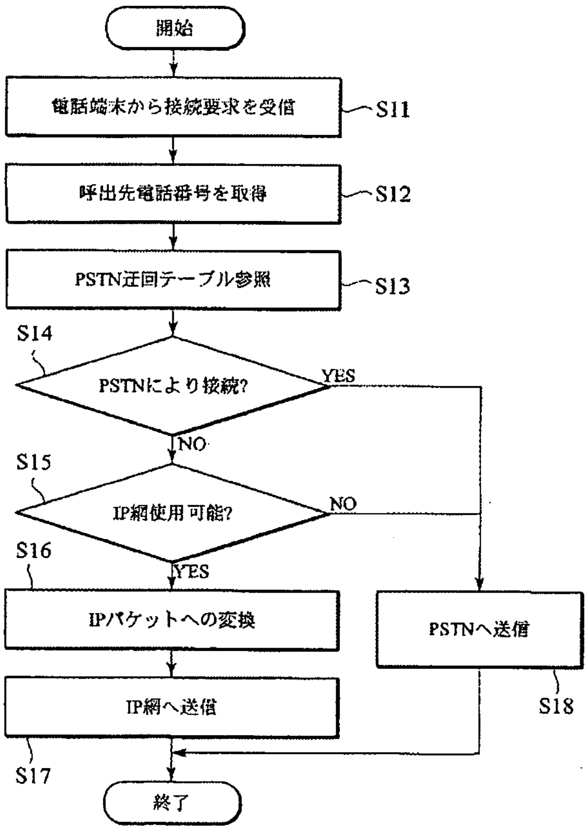
[Drawing 5]

FIG. 5



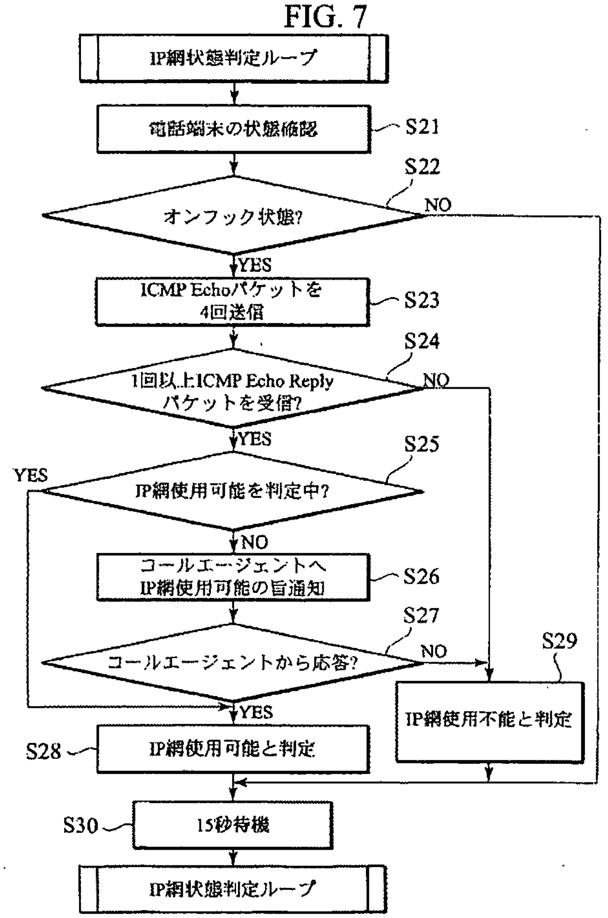
[Drawing 6]

FIG. 6



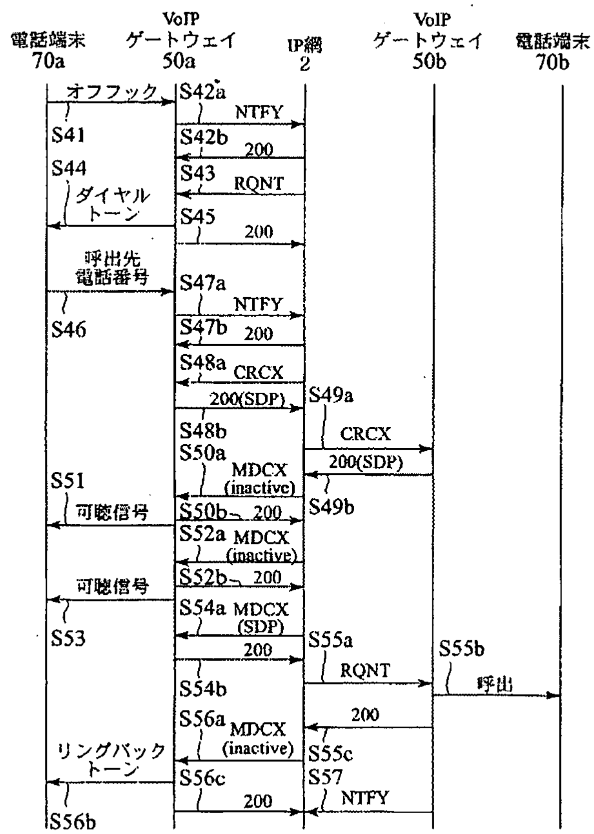
[Drawing 7]

FIG. 7



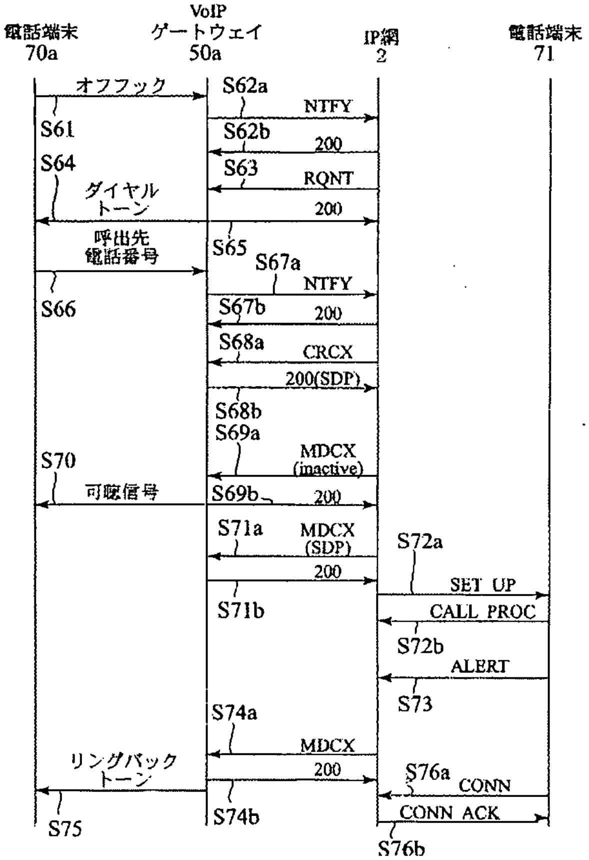
[Drawing 8]

FIG. 8



[Drawing 9]

FIG. 9



[Drawing 10]

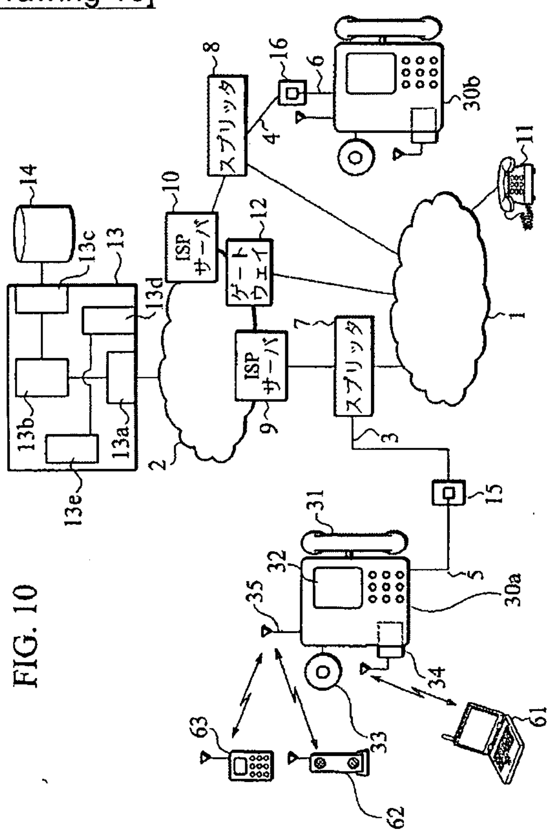
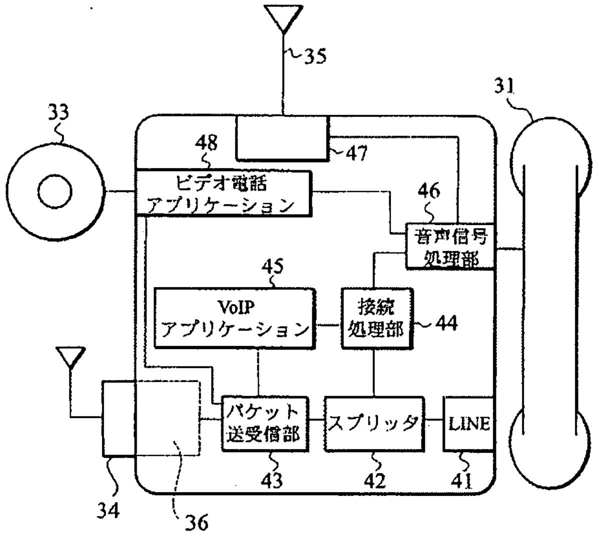


FIG. 10

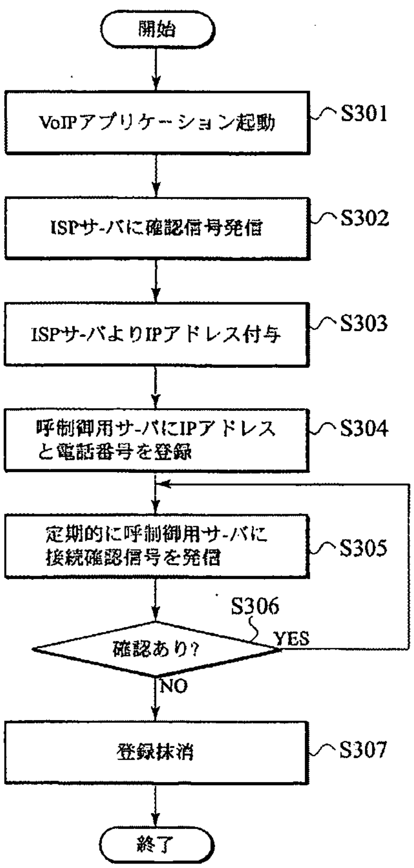
[Drawing 11]

FIG. 11



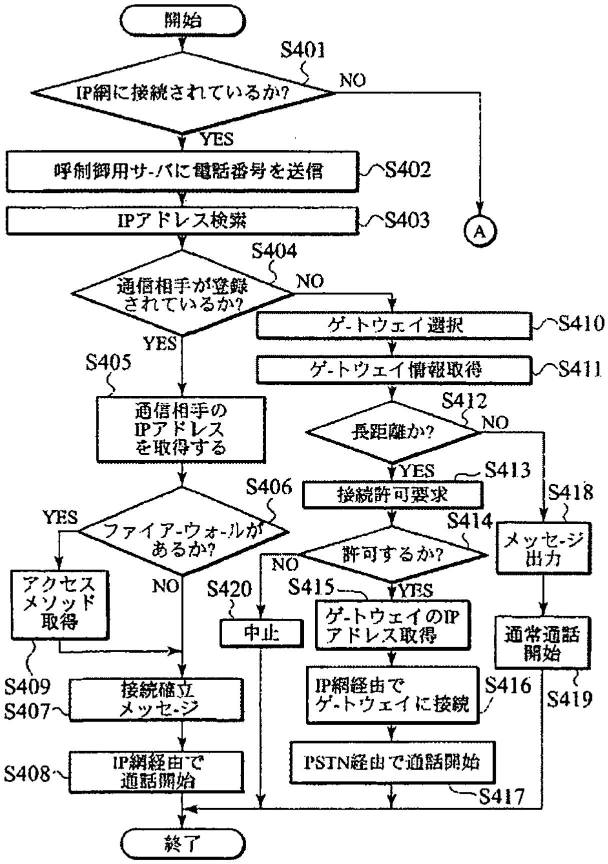
[Drawing 12]

FIG. 12



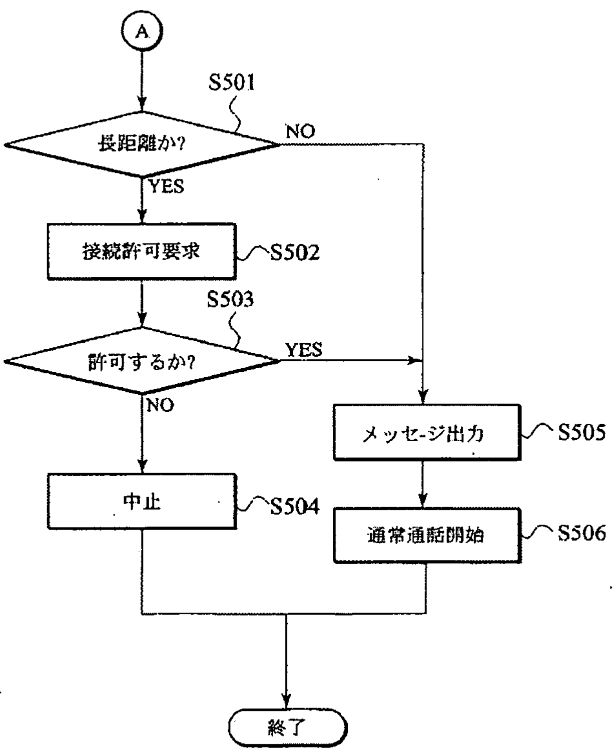
[Drawing 13]

FIG. 13



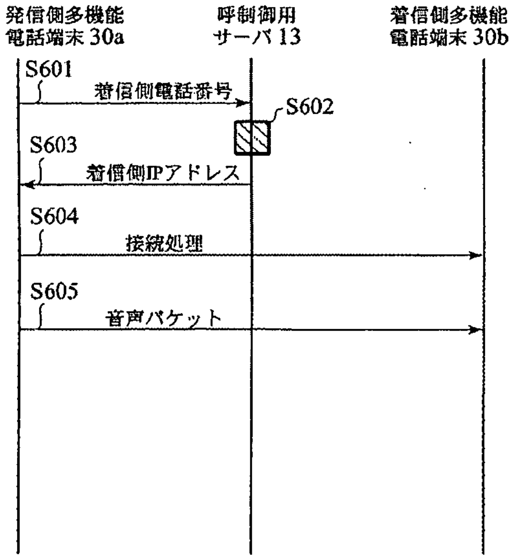
[Drawing.14]

FIG. 14



[Drawing 15]

FIG. 15



[Drawing 16]

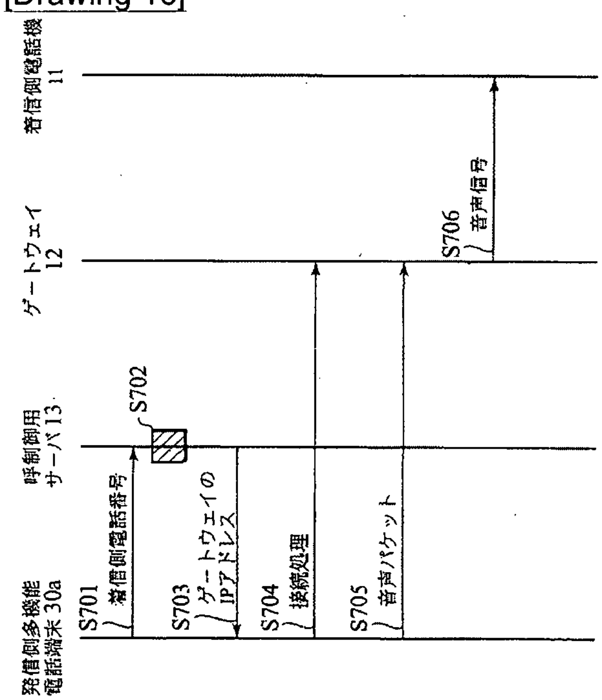


FIG. 16

[Translation done.]